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Performance Analysis Of Voip Using Udp And Tcp Protocol

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Abstract— **This Voice over IP is a technology which consists of sending voice over the IP heterogeneous network Due to the recent technological advancement in packet switching networks, VoIP (Voice over Internet Protocol) has become an industry favourite over Public Switching Telephone Networks (PSTN) with regards to voice communication. The cheap cost of making a call through VoIP internationally and paying just one bill for data usage is a huge benefit.**

Although, there are huge advantages to using VoIP, there are some known issues such as, packet loss, jitter and latency due to the connection through the computer internet. The main purpose of this work consists of studying these problematic and to create a VoIP network with NS 2.35 and testing for its known faults. Through this we want to get a better understanding of the underlying layers of the network (TCP and UDP) and see if and where improvements can be made..

***Index Terms*—TCP, UDP, VoIP, Network**

I. Introduction

VoIP applications with a strong time pressure are not supported by the Internet. The service offered by the latter does not offer a guarantee in terms of quality of service. Although widely used, the TCP protocol does not solve all the constraints of the voice due to his lack of vis-à-vis flexibility of latency. All these reasons mean that IP telephony should know dynamically operate the connectionless UDP offered. To measure the performance of VoIP vis-a-vis these two protocols, performance parameter measurements such as packet loss, jitter, latency and throughput are needed. These points will be discussed in this paper.

II. Methodology

A. Goal of the simulation

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B. Final Stage

When The simulation aims to compare the performance of VoIP deployed TCP and UDP. It is to recreate reality in a manner consistent VoIP deployment in both cases.

By simulating network conditions, protocols, codecs, and volumes of data generated by the application, it is possible to have a comprehensive vision on the performance of VoIP on different network characteristics.

Our simulation will be:

Designing a system model (real) studied

Evaluate the parameters of VoIP performance (packet loss, throughput, jitter, latency (time from start to finish)). Put in the form of graphs the results of the simulation.

C. FSimulation of VoIP in NS-2:scenario.

In our implementation, we will use the circuit switching principle between the transmitter and receiver.In the shifting standard circuits, the analog voice signal must be sampled much as twice the bandwidth. Standardization called MIC (Many Integrated Core) application 8000 samples per second, and 8 bits per sample, there will therefore be 8-bit all 125 µs, Then we will use a bandwidth of 64 kb / s for voice. The integration of a G.711 codec will emulate this constraint throughput and bandwidth. Furthermore, the size of transmitted packets is chosen to be 128 bytes for UDP, 1040 bytes for the payload and 40 bytes for TCP acknowledgment (ACK).The following table (Table1) summarizes this configuration:

Table 1:Packet size

Type	size
UDP	128 octets
TCP	1040 octets
ACK	40 octets

According to the figure below Figure 1, our scenario is a telephone communication (VoIP) between nodes 0 and 4. To properly simulate a call, the choice of an exponential type of traffic has been done. This will allow us to emulate a typical conversation that includes being averaged according to an exponential law. The node 0 is active for a while "on" every 120 ms. This time is devoted to the transmission of data (words). It is also provided with an idle time (idle time) every 88 ms. This simulates the time that Node A Node B listen for a while. The node 4 sends fewer packets on an average uptime of 88 ms and a period of inactivity corresponding to 128 ms. These different values are the inverse of the node 0. We assume that the idle time is when the user listens to his interlocutor.

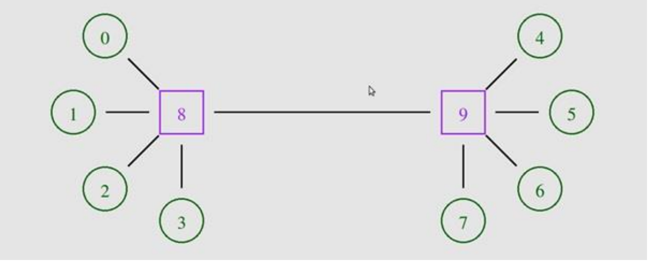

Fig 1: Configuration of our VoIP simulation wired

Furthermore, it is important to note that the simulation scale is chosen to emulate a VoIP communication between two companies located over a distance of tens of kilometers. The connection between the routers 8 and 9 is a full duplex connection to a bandwidth of 5.92 Mb / s.In this figure, we can see the two locations that communicate and are connected

by a wired Ethernet support.Moreover, to simulate the ADSL technology which is designed to combine voice, and other data (video), we will associate the other background traffic in parallel to the voice (background traffic). These cross traffic is exchanged between the nodes: 1, 2.3, 5, 6, and 7 with a constant bit rate.

At t = 0, the simulation begins with the launch of VoIP flows between node 0 and the node 4. In parallel, the nodes 1 and 5 create a constant bit rate of background traffic (CBR) at a rate of 5.89 Mb / s each (the other nodes are idle).

This traffic will provide a load close to that supported the duplex link between the two routers (8 and 9). We expect then to see that queue of each of the two routers is full.

Then, between 2 seconds and 4 seconds (simulation scale), nodes 2 and 6 began their exchange of packets (previously connected nodes 1 and 5 are off) with a background traffic rate 5.91 Mb / s each. This value is chosen to create a router overload, creating a first network congestion.

Then, at time t = 4 s, nodes 3 and 7 begin their communication with background traffic rate of 5.92 Mb / s each, which far exceeds the bandwidth of the network. At this stage, it is expected that the queue in the two routers is full, which will result in a large number of lost packets.

In this simulation, UDP and TCP agents are attached to the nodes 0 and 4. For cross-traffic, we use UDP-based constant bit rates.

By applying this algorithm in a terminal linux (Ubuntu). We will have a new log file that is much more optimal capacity.

III. Processing with Matlab.

imageimageMatlabCurveControl awkimageFile tracefile out.tr

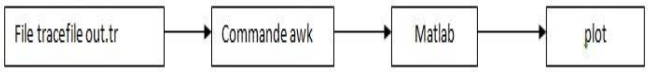

Fig2 Matlab processing plot

Because the simulation was performed with a Unix platform, it is then necessary to transpose the Matlab software on this open source operating system.

This transposition can sometimes be tedious because this software is not free for use on all platforms. Indeed, to date, Matlab works only with the Windows operating system.

Consequently, an alternative is to install his counterpart who is the package track graph.

IV. Flow rate

From what we saw in the previous chapter, in the case of transmitting data from a node A to a node B, flow data consists of numbers measured in bytes / second received by event B. Type "r" in the first column of the trace file is the main argument used by Matlab and Trace graph to calculate the flow rate.

V. Packet loss

To calculate the packet loss, it is usually based on the number of packets that were discarded between the path from transmitter to receiver, that is to say between node 0 and node 4. Two measures were taken, one describes the instantly lost packet which describes how many packets are lost on each time slot, then the other measurement describes the cumulative number that describes the total number of lost packets during the simulation.

The event type "d" in the trace is the main element for the determination of this second type of performance parameter.

VI. Period-end (end to end delay)

This performance parameter is the time it takes a packet to travel the path between node A and node B. It is measured according to the following. To get its value, it should follow the steps below.

To obtain the time when the packet is created or transmitted.

Get the time when the same packet arrives at destination.

Take the time difference between the two events.

To implement this procedure, at first we seek the time when the packet is created by checking the trace file, the type of event "+" on the relevant node. Then, the unique identifier specified in column 12 must be saved and a new search of the same identifier coupled with type of event "r" is made. The time to look is then obtained by taking the difference of the times found.

VII. Jitter:

Jitter is also known as IP Packet Delay Variation name (IPDV) which means packet delay variation is defined as the difference of time it takes to travel between the two ends of the communication. Specifically, there are several ways to calculate this parameter. The simplest is shown by the following formula.

A $$G = \frac{TRPA - TRDP}{Diff\,(seq\ A - seqB)}$$

TRPA: reception time of the current package.

TRDP: the last packet reception time.

Diff (seq seq A- B) is the difference in the number sequence between two packages.

VIII. VoIP simulation with UDP

UDP is a simple protocol that sends data from the application layer to the IP layer for transmission. It does not check and therefore the flow of data is not assured. A UDP header is simply an optional source port, destination port, the length of the datagram, and a checksum.

The main reason for the deployment of voice over UDP is to optimize the latency (time from start to finish). In general, sporadic packet loss in a conversation is not disturbing with a loss of 5%.The figure below shows the NAM window at the beginning of the simulation is to say, cross the first traffic coupled with the VoIP flows. The colors of traffic are chosen to show their difference in terms of speed and be able to better visualize how is network congestion.

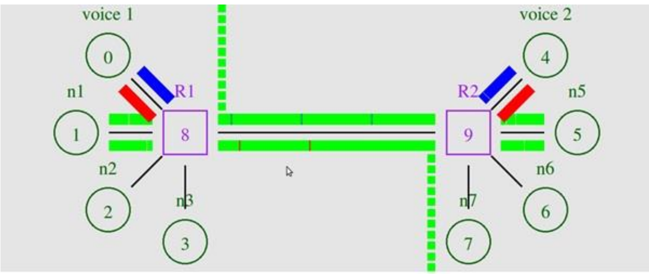

Fig3 VoIP simulation using UDP in NAM - first substantive traffic

A. *Parameter Rating: Rate*

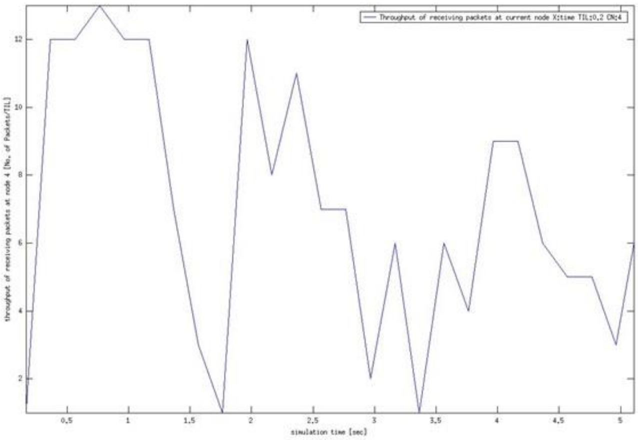

Fig4 Node Flow 0

Initially, we see a rate of 64 kb / s for the voices of the two ends (node 0 and node 4). And then with the parallel cross the first traffic (Fig4, we see a rate decrease of both ends

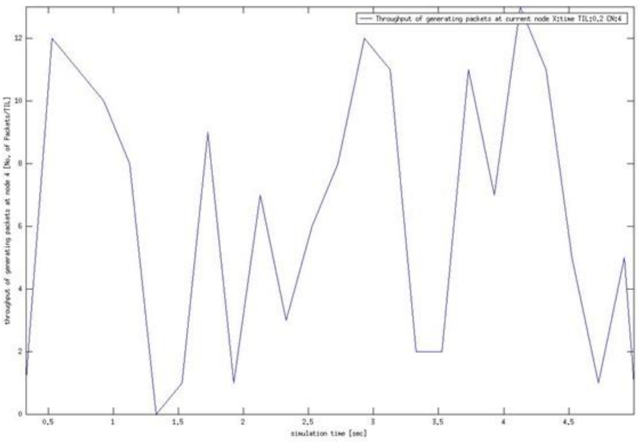

Fig4 Node Flow 4

For node 0, between [1.5; 2] and [3; 3.5], cross traffic takes a big portion of the bandwidth, which decreases at lower throughput. However after the descent of times, we are seeing an increase in throughput is explained by the fact that the node 0 does not speak during these periods and increasing throughput partially.

As for the one node 4 (Fig5), the lowest rate is just before t = 1.5, which also explains the weight cross traffic that takes full extent of bandwidth. We are also seeing increases in rates (equivalent to the initial rate) that explain the disparity of the idle time of the node 4, which tends more to listen to the voice of the node 0 as send.

B. *Parameter Rating: period-end*

For UDP, the time on both sides are around 90 ms. That is to say, well below the value of the recommended limit is 150 ms. This delay is unacceptable for VoIP communication. We also note that this period varies depending on background traffic that occurs.

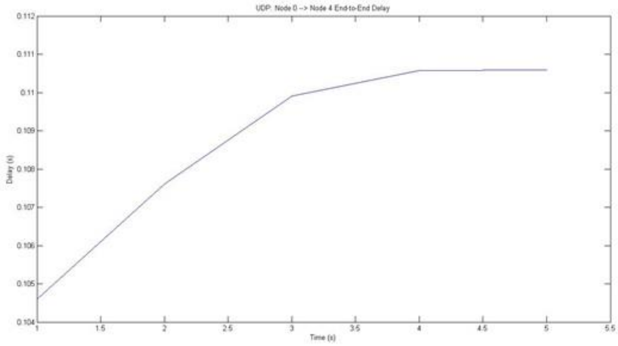

Fig 6 Period-end node 0 -> node 4 (UDP)

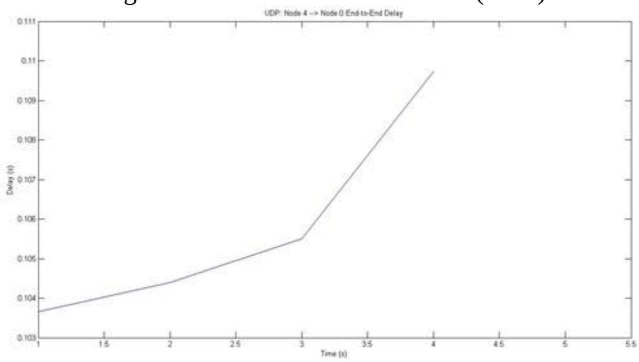

Fig7 Period-end node 4 -> node 0 (UDP)

C. *Parameter Rating: packet loss*

Generally, packet loss begins to be visible at t = 2.5 s, this is due to the sudden congestion caused by the combination of background traffic and the two speakers speaking simultaneously. At t = 3 s, the packet loss has increased and will not stop until the end of communication. This deterioration follows the appearance of an exponential curve as the bandwidth occupied by background traffic continues to increase. Indeed, the queue fills up too quickly, causing the release of several packages, as background traffic flow with a constant bit rate, the queues on both routers (8 and 9) have no time to process all packets passing through it by them. In the figure below (Figure 5.09), we see all the lost packets during the simulation (UDP packet, VoIP packet).

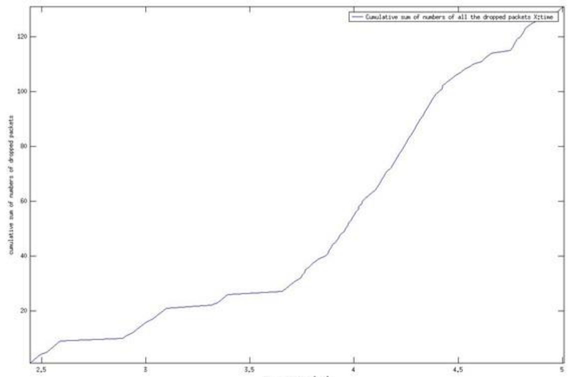

Fig8 Total cumulative packet loss of all nodes in the simulation (UDP).

Furthermore, in the figure below , the left represents the packet loss from node 0 and the right represents the node 4. Typically the reasons for these losses are the same as before. Nevertheless, it should be emphasized that the node 0 starts earlier to lose packets that the node 4. This is because the idle time that is uneven and is more concentrated at the node 4. The node 0 sends more data first, the node 4 only listens during this period.

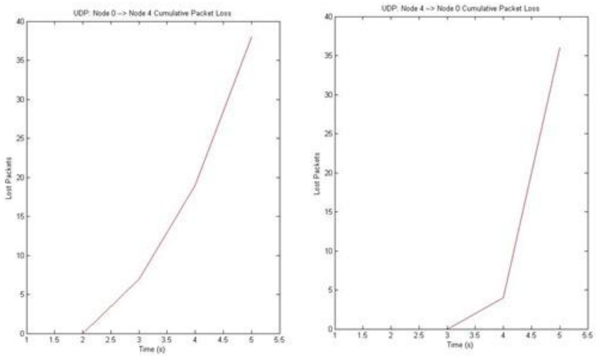

Fig9 Loss identified at the node 0 and node 4 (UDP)

D. Evaluation of the parameter: Gigue.

In both curves, we can conclude that there is a minimal delay variation from end to end between nodes. What is correct for VoIP.

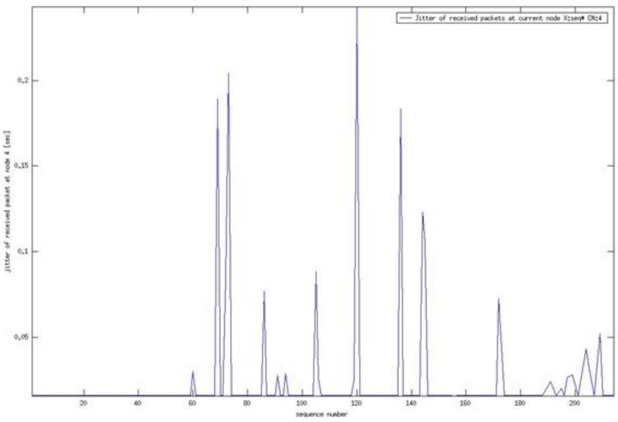

Fig10 Jitter node 0 (UDP)

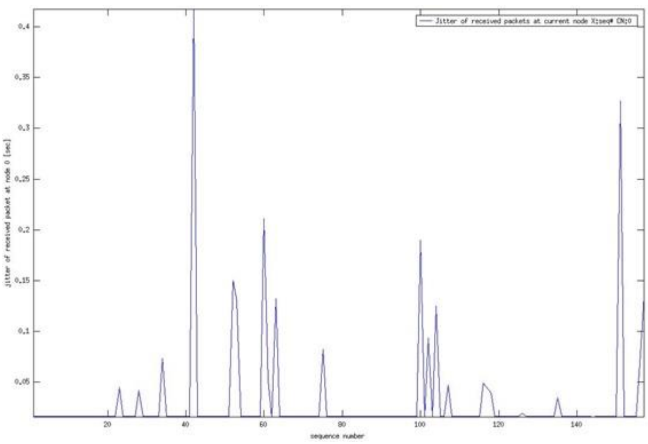

Fig11 Jitter node 4 (UDP)

IX. VOIP SIMULATION WITH TCP

The IP protocol is based on the principle of best-effort to route packets. TCP comes as reinforcements to mitigate this error by detecting errors and retransmitting lost packets.The TCP transport protocol is connection-oriented is to say that a communication must always be acknowledged by an ACK. However, despite these services, its drawback is the excessive delay. We will see this in more detail in this second simulation case. The following figure shows an illustration of this emulator (Fig12).

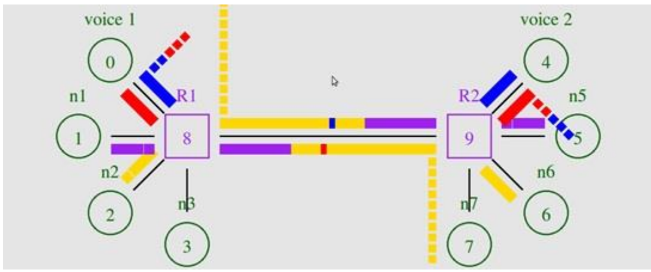

Fig 12 VoIP simulation using TCP

A. Parameter Rating: Rate

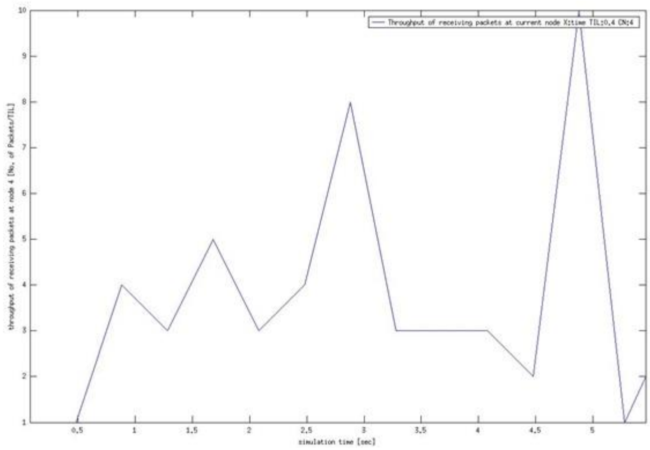

Fig 13 0 node throughput (TCP)

It can be seen that the two curves (Fig12,13) are quite similar except in the period [4.5; 5], the scale of 1 (ordered axis) means a rate of 64 kb / s.This rate is respected on average throughout the simulation. We can say that there is no loss of performance in both sides.

However, there is some flow drops at t = 0.5 and t = 1.5. These decreases can be explained first by network congestion by routers 8 and 9. Indeed, ACK packets also take bandwidths depriving useful information to cross the wire carrier in the best conditions.

Then these declines are caused by the ramp-up of the flow of some background traffic. The medium is shared to best effort, which means cuts the flow of both sides.

Furthermore, as regards peaks that lie over the entire curve, this is caused by congestion control. Indeed thanks to Slow Start and Congestion Avoidance mechanisms present in TCP. The flow progresses through successive stages. Then when a bearing is not properly validated because the acquittals of frames sent from reaching over to the transmitter, the flow is

automatically lowered.

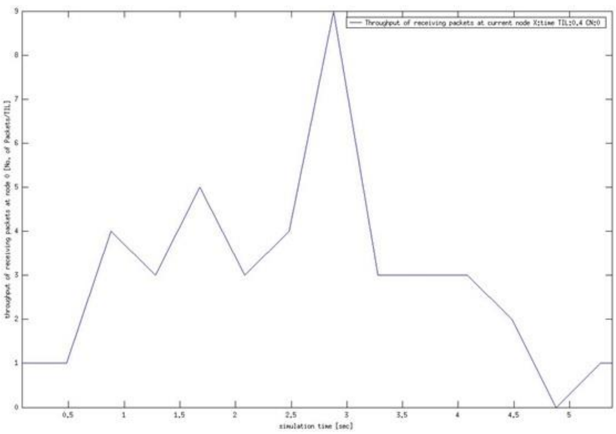

Fig 14 Node Flow 4 (TCP)

B. Rating setting: Delay from start to finish.

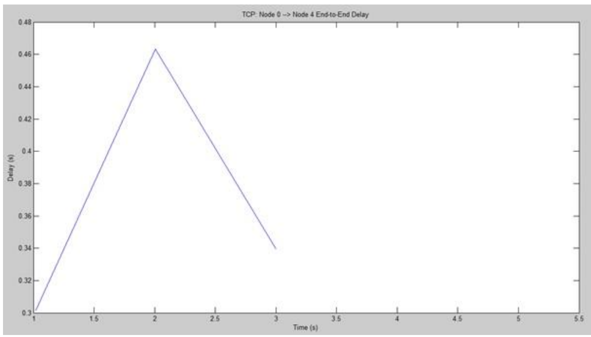

Fig15;Period-end node 0 (TCP)

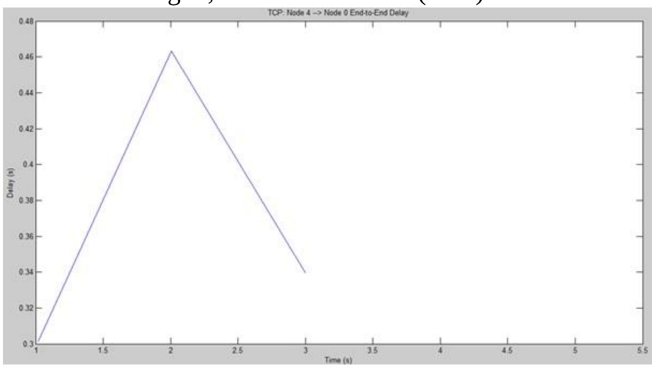

Fig16 Period-end node 4 (TCP)

On these two curves (Fig.15 and Fig16), we see significant delays ranging from 0.3 s (300 ms) to 0.5 s (500 ms). These values exceed not only the recommendation of 150 ms but also that of 400 ms (international communication).The reason for the significant delay is caused by the error control. Indeed each receiver node sends an acknowledgment message for all received frames. This may be cumulative to pay more frames simultaneously. The side effect is that the sender must always wait for an acknowledgment to send the next packet.

C. Evaluation of the parameter: Loss of packages.

In both figures (Fig15,16), the upper figure represents the instantaneous and cumulative losses of packets for Node 0 and that of the right is that of the node 4.

First, for the node 0, we find that it is 4 s we record a loss of voice packets. Before this time, the node 0 loses no package. This performance is gained through the flow control and error control. Then, after 4 seconds, we begin to lose the voice data since not only the background traffic sent to a similar rate to the rate link is 5.92 Mb / s, but also because the previous packet background traffic are not all sent. Therefore, congestion will be provided on the side of two routers, which reject many packages

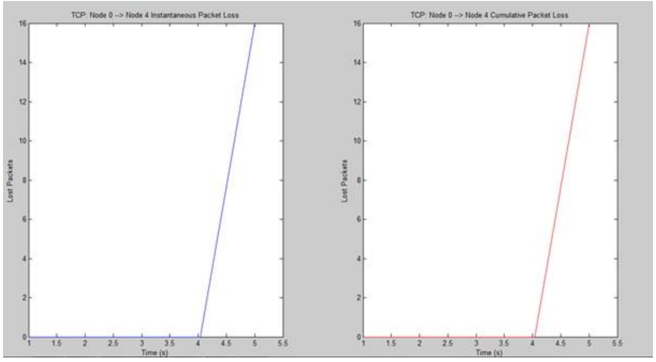

Fig17 Packet loss from node 0

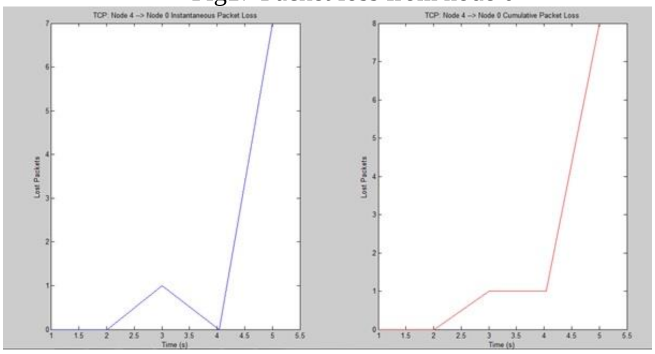

Fig18 Losses packages node 4.

Second, to node 4, we start seeing losses earlier, generally ACK packets, ACK loss reflects his downtime. Indeed when active, the network is congested by traffic means, however, at this stage of ACK losses do not disturb the communication. Then as node 0, it begins to lose a lot of packets because of network congestion.

X. Conclusion

we were able to experience our comparative study by simulation under NS-2.

The phone remains one of the dominant applications of the network world, for many years to come, mainly because of the emergence of new and huge markets. However, the quality is very variable depending on efforts made by the corporate network operators and telecommunications network operators.The issues are numerous and sometimes complex. IP telephony is not a simple application to implement in the context of the integration of all telecommunications services on the same network.At the end of this memory, we saw that the performance of VoIP is measurable from performance parameters which are packet loss, audio codec, latency (time from start to finish) jitter and throughput . The measurement of these parameters allowed us to evaluate the performance of VoIP technology. We have seen that VoIP is not compatible with the standard Internet TCP transport protocol. He adds further delays that induce a data reference. As a result, VoIP then uses the UDP protocol, which provides more

flexibility.In our simulation, we validated the UDP fact gives the best result in terms of transport protocol. Indeed, despite that we lose much of UDP packets than TCP, we still favorable results for latency and jitter that are more sensitive in the VoIP packet loss. A packet loss of 5% is still acceptable effect.

However, a possible viable solution would always be based with TCP through improvements in vis-à-vis transport of multimedia applications. The candidate Protocol looming at the moment is the transport protocol SCTP (Stream Control Transmission Protocol). It is a new protocol that is not yet mature, but still deserves to be confronted with the integration of VoIP. An implementation of VoIP test under SCTP should be run in the future.

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